

Lab 3 - RTP Header Compression

CCNA Voice Lab 3

RTP Header Compression

Lab Objective:

The objective of this lab exercise is for you to learn and understand how implement Compressed RTP on Cisco IOS H.323 gateways

Topic Description:

Compressed RTP (cRTP) reduces the RTP header to 2 or 4 bytes. It is used to save bandwidth on WAN links in VoIP networks

Lab Difficulty:

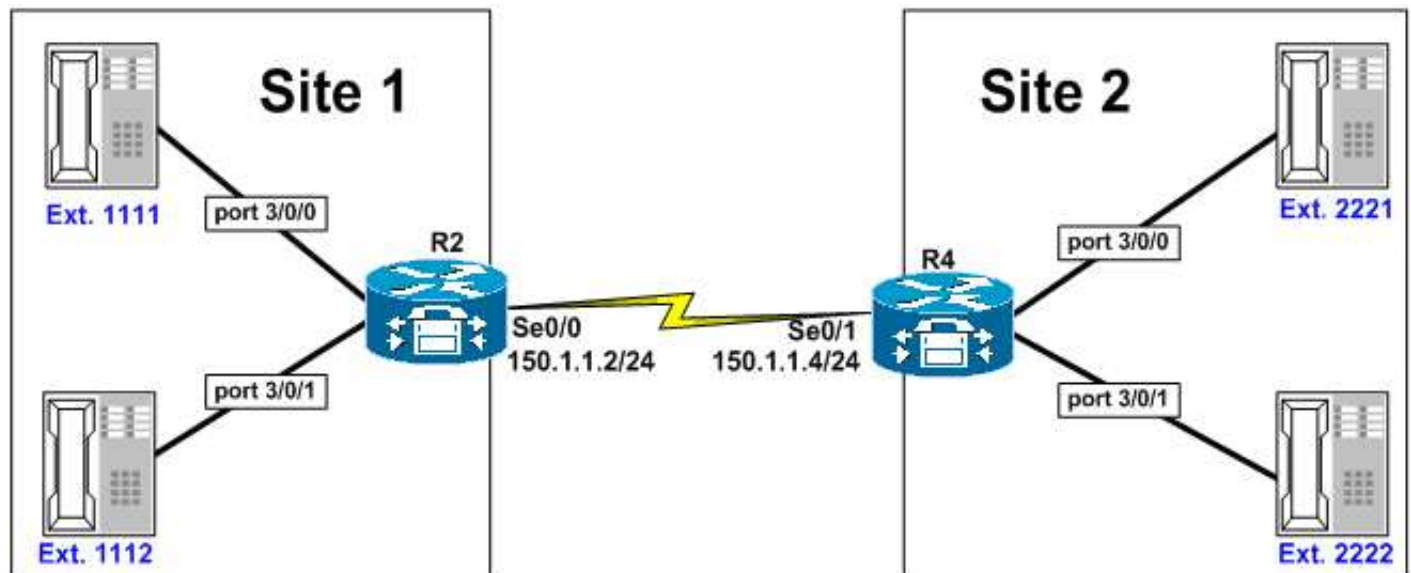
This lab has a difficulty rating of 5/10

Readiness Assessment:

When ready for your exam, you should be able to complete this lab in no more than 10 minutes

Lab Topology:

Please use the following topology to complete this lab exercise:



Lab 3 Configuration Tasks

Task 1:

Configure the hostnames and IP addresses on R2 and R4 as illustrated in the diagram. Configure R4 to provide clocking to R2. Configure the clock rate on R4 as 800Kbps

Task 2:

Configure POTS dial peers on R2 using the information in the diagram. You are permitted to use any dial peer numbers that you want to use

Task 3:

Configure POTS dial peers on R4 using the information in the diagram. You are permitted to use any dial peer numbers that you want to use

Task 4:

Configure a single VoIP dial peer on R2 to the R4 extension range. You are permitted to use any dial peer numbers that you want to use. Use SIP as the signaling protocol for the dial peer

Task 5:

Configure a single VoIP dial peer on R4 to the R2 extension range. You are permitted to use any dial peer numbers that you want to use. Use SIP as the signaling protocol for the dial peer

Task 6:

Configure cRTP on R2 so that it actively compresses all RTP packets

Task 7:

Configure cRTP on R4 so it compresses RTP packets only if it receives compressed packets

Task 8:

Verify your configuration using the appropriate Cisco IOS **show** commands

Lab 3 Configuration and Verification

Task 1:

```
R2(config)#hostname R2
R2(config)#interface Serial0/0
R2(config-if)#description .Connected To R4 Serial 0/1.
R2(config-if)#ip address 150.1.1.2 255.255.255.0
```

```
R4(config)#hostname R4
R4(config)#interface Serial0/1
R4(config-if)#description .Connected To R2 Serial 0/0
R4(config-if)#ip address 150.1.1.4 255.255.255.0
R4(config-if)#clock rate 800000
```

Verify connectivity by performing a simple ping between the two routers

```
R2#ping 150.1.1.4
```

Type escape sequence to abort.

Sending 5, 100-byte ICMP Echos to 150.1.1.4, timeout is 2 seconds:

!!!!

Success rate is 100 percent (5/5), round-trip min/avg/max = 4/4/4 ms

```
R4#ping 150.1.1.2
```

Type escape sequence to abort.

Sending 5, 100-byte ICMP Echos to 150.1.1.2, timeout is 2 seconds:

!!!!

Success rate is 100 percent (5/5), round-trip min/avg/max = 4/4/8 ms

Task 2:

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#destination-pattern 1111
R2(config-dial-peer)#port 3/0/0
R2(config-dial-peer)#exit
R2(config)#dial-peer voice 2 pots
R2(config-dial-peer)#destination-pattern 1112
R2(config-dial-peer)#port 3/0/1
R2(config-dial-peer)#exit
```

Task 3:

```
-----  
R4(config)#dial-peer voice 1 pots  
R4(config-dial-peer)#destination-pattern 2211  
R4(config-dial-peer)#port 3/0/0  
R4(config-dial-peer)#exit  
R4(config)#dial-peer voice 2 pots  
R4(config-dial-peer)#destination-pattern 2221  
R4(config-dial-peer)#port 3/0/1  
R4(config-dial-peer)#exit
```

Task 4:

```
R2(config)#dial-peer voice 3 voip  
R2(config-dial-peer)#destination-pattern 222.  
R2(config-dial-peer)#session target ipv4:150.1.1.4  
R2(config-dial-peer)#session protocol sipv2  
R2(config-dial-peer)#exit
```

Task 5:

```
R4(config)#dial-peer voice 3 voip  
R4(config-dial-peer)#destination-pattern 111.  
R4(config-dial-peer)#session target ipv4:150.1.1.2  
R4(config-dial-peer)#session protocol sipv2  
R4(config-dial-peer)#exit
```

Task 6:

```
R2(config)#int s0/0  
R2(config-if)#ip rtp header-compression  
R2(config-if)#exit
```

Task 7:

```
R4(config)#int s0/1  
R4(config-if)#ip rtp header-compression passive  
R4(config-if)#exit
```

Task 8:

The first validation task is to place a call from a Site 1 extension to a Site 2 extension:

```
R2#show sip-ua calls  
SIP UAC CALL INFO
```

```
Call 1  
SIP Call ID      : F6A08185-2BE111D6-8008A096-5388F491@150.1.1.2  
State of the call : STATE_ACTIVE (7)  
Substate of the call : SUBSTATE_NONE (0)  
Calling Number : 1111  
Called Number : 2222  
Bit Flags : 0x12120030 0x100000 0x400  
CC Call ID : 3  
Source IP Address (Sig ): 150.1.1.2  
Destn SIP Req Addr:Port : 150.1.1.4:5060  
Destn SIP Resp Addr:Port: 150.1.1.4:5060  
Destination Name : 150.1.1.4  
Number of Media Streams : 1  
Number of Active Streams: 1  
RTP Fork Object : 0x0  
Media Stream 1  
State of the stream : STREAM_ACTIVE  
Stream Call ID : 3  
Stream Type : voice-only (0)  
Negotiated Codec : g729r8 (20 bytes)  
Codec Payload Type : 18  
Negotiated Dtmf-relay : inband-voice  
Dtmf-relay Payload Type : 0
```

Media Source IP Addr:Port: 150.1.1.2:18310
Media Dest IP Addr:Port : 150.1.1.4:16844
Orig Media Dest IP Addr:Port : 0.0.0.0:0

Number of SIP User Agent Client(UAC) calls: 1

SIP UAS CALL INFO

Number of SIP User Agent Server(UAS) calls: 0

Next, verify that RTP traffic is being compressed. To generate RTP traffic, you must have physical access to the phone and talk, etc, to send packets across the WAN. If you do not have physical access to the phone, simply use this as a command verification exercise

R2#show ip rtp header-compression

RTP/UDP/IP header compression statistics:

Interface Serial0/0 (compression on, Cisco, RTP)

Rcvd: **6980 total, 6978 compressed**, 0 errors, 0 status msgs

0 dropped, 0 buffer copies, 0 buffer failures

Sent: **6853 total, 6851 compressed**, 0 status msgs, 42 not predicted

259996 bytes saved, 150887 bytes sent

2.72 efficiency improvement factor

Connect: 16 rx slots, 16 tx slots,

2 misses, 0 collisions, 0 negative cache hits, 15 free contexts

99% hit ratio, five minute miss rate 0 misses/sec, 0 max

The same can be verified on R4 as well:

R4#show ip rtp header-compression

RTP/UDP/IP header compression statistics:

Interface Serial0/1 (compression on, Cisco, RTP, passive)

Rcvd: **7036 total, 7032 compressed**, 0 errors, 0 status msgs

0 dropped, 0 buffer copies, 0 buffer failures

Sent: **9251 total, 9249 compressed**, 0 status msgs, 23 not predicted

351276 bytes saved, 203631 bytes sent

2.72 efficiency improvement factor

Connect: 16 rx slots, 16 tx slots,

2 misses, 0 collisions, 0 negative cache hits, 15 free contexts

99% hit ratio, five minute miss rate 0 misses/sec, 0 max

Final Relevant Configuration R2:

R2#show running-config

Building configuration...

Current configuration : 1568 bytes

!

version 12.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname R2

!

!

!

!

interface Serial0/0

description .Connected To R4 Serial 0/1.

ip address 150.1.1.2 255.255.255.0

ip rtp header-compression

!

!

!

!

voice-port 3/0/0

|

```

.
voice-port 3/0/1
!
voice-port 3/1/0
!
voice-port 3/1/1
!
!
!
!
dial-peer voice 1 pots
 destination-pattern 1111
 port 3/0/0
!
dial-peer voice 2 pots
 destination-pattern 1112
 port 3/0/1
!
dial-peer voice 3 voip
 destination-pattern 222.
 session protocol sipv2
 session target ipv4:150.1.1.4

```

Final Relevant Configuration R4:

R4#**show running-config**

Building configuration...

Current configuration : 1571 bytes

```

!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname R4
!
!
!
!
interface Serial0/1
 description .Connected To R2 Serial 0/0
 ip address 150.1.1.4 255.255.255.0
 clock rate 800000
 ip rtp header-compression passive
!
!
!
!
voice-port 3/0/0
!
voice-port 3/0/1
!
voice-port 3/1/0
!
voice-port 3/1/1
!
!
!
!
dial-peer voice 1 pots
 destination-pattern 2221
 port 3/0/0
!
dial-peer voice 2 pots
 destination-pattern 2222
 port 3/0/1

```

port 5060/1

!

dial-peer voice 3 voip

destination-pattern 111.

session protocol sipv2

session target ipv4:150.1.1.2



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